(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) EP 0 899 720 A2

(12) EUROPEAN PATENT APPLICATION

(43) Date of publication:
03.03.1999 Bulletin 1999/09

(51) Int Cl.[6]: **G10L 9/14**

(21) Application number: 98306906.3

(22) Date of filing: 27.08.1998

(84) Designated Contracting States:
AT BE CH CY DE DK ES FI FR GB GR IE IT LI LU MC NL PT SE
Designated Extension States:
AL LT LV MK RO SI

(30) Priority: 28.08.1997 US 57114 P

(71) Applicant: **TEXAS INSTRUMENTS INC.**
Dallas, Texas 75243 (US)

(72) Inventor: **McCree, Alan V.**
Dallas Texas 75248 (US)

(74) Representative: **Potter, Julian Mark et al**
D. Young & Co.,
21 New Fetter Lane
London EC4A 1DA (GB)

(54) **Quantization of linear prediction coefficients**

(57) A new method for quantization of the LPC coefficients in a speech coder includes a new weighted error measure including every frame sampling an impulse response from LPC filter (21) of said coder, filtering the samples using a perceptual weighting filter (47) and processing in a computer (39) to calculate autocorrelation function of the weighted impulse response, computing Jacobian matrix for LSF (Line Spectral Frequency), computing correlation of rows of Jacobian matrix and calculating LSF weights by multiplying correlation matrices.

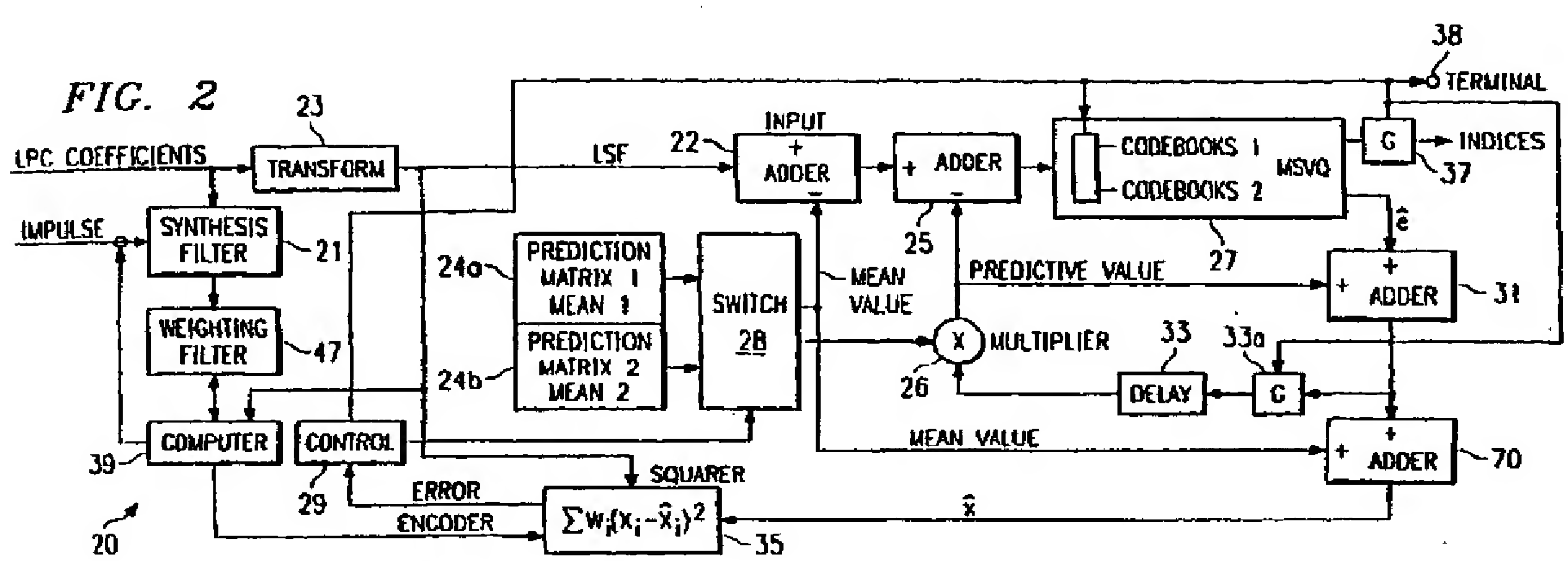


FIG. 2

EP 0 899 720 A2

Description

TECHNICAL FIELD OF THE INVENTION

[0001] This invention relates to switched-predictive vector quanzization and more particularly to quantization of LPC coefficients transformed to line spectral frequencies.

BACKGROUND OF THE INVENTION

[0002] Many speech coders, such as the new 2.4 kb/s Federal Standard Mixed Excitation Linear Prediction (MELP) coder (McCree, et al., entitled, "A 2.4 kbits/s MELP Coder Candidate for the New U. S. Federal Standard," Proc. ICASSP-96, pp. 200-203, May 1996.) use some form of Linear Predictive Coding (LPC) to represent the spectrum of the speech signal. A MELP coder is described in the Applicant's co-pending Application Serial No. 08/650,585, entitled "Mixed Excitation Linear Prediction with Fractional Pitch," filed 05/20/96, incorporated herein by reference. Fig. 1 illustrates such a MELP coder. The MELP coder is based on the traditional LPC vocoder with either a periodic impulse train or white noise exciting a 10th order on all-pole LPC filter. In the enhanced version, the synthesizer has the added capabilities of mixed pulse and noise excitation periodic or aperiodic pulses, adaptive spectral enhancement and pulse dispersion filter as shown in Fig. 1. Efficient quantization of the LPC coefficients is an important problem in these coders, since maintaining accuracy of the LPC has a significant effect on processed speech quality, but the bit rate of the LPC quantizer must be low in order to keep the overall bit rate of the speech coder small. The MELP coder for the new Federal Standard uses a 25-bit multi-stage vector quantizer (MSVQ) for line spectral frequencies (LSF). There is a 1 to 1 transformation between the LPC coefficients and LSF coefficients.

[0003] Quantization is the process of converting input values into discrete values in accordance with some fidelity criterion. A typical example of quantization is the conversion of a continuous amplitude signal into discrete amplitude values. The signal is first sampled, then quantized.

[0004] For quantization, a range of expected values of the input signal is divided into a series of subranges. Each subrange has an associated quantization level. A sample value of the input signal that is within a certain subrange is converted to the associated quantizing level. For example, for 8-bit quantization, a sample of the input signal would be converted to one of 256 levels, each level represented by an 8-bit value.

[0005] Vector quantization is a method of quantization, which is based on the linear and non-linear correlation between samples and the shape of the probability distribution. Essentially, vector quantization is a lookup process, where the lookup table is referred to as a "codebook". The codebook lists each quantization level, and each level has an associated "code-vector". The vector quantization process compares an input vector to the code-vectors and determines the best code-vector in terms of minimum distortion. Where x is the input vector, the comparison of distortion values may be expressed as:

$$d(x, y^{(j)}) < d(x, y^{(k)}),$$

for all j not equal to k. The codebook is represented by $y^{(j)}$, where $y^{(j)}$ is the jth code-vector, $0 < j < L$, and L is the number of levels in the codebook.

[0006] Multi-stage vector quantization (MSVQ) is a type of vector quantization. This process obtains a central quantized vector (the output vector) by adding a number of quantized vectors. The output vector is sometimes referred to as a "reconstructed" vector. Each vector used in the reconstruction is from a different codebook, each codebook corresponding to a "stage" of the quantization process. Each codebook is designed especially for a stage of the search. An input vector is quantized with the first codebook, and the resulting error vector is quantized with the second codebook, etc. The set of vectors used in the reconstruction may be expressed as:

$$y^{(j_0,j_1,\cdots j_{S-1})} = y_0^{(j_0)} + y_1^{(j_1)} + y^{(j_{S-1})_{S-1}}$$

, where S is the number of stages and y_s is the codebook for the sth stage. For example, for a three-dimensional input vector, such as x = (2,3,4), the reconstruction vectors for a two-stage search might be y_0 = (1,2,3) and y_1 = (1,1,1) (a perfect quantization and not always the case).

[0007] During multi-stage vector quantization, the codebooks may be searched using a sub-optimal tree search algorithm, also known as an M-algorithm. At each stage, M-best number of "best" code-vectors are passed from one stage to the next. The "best" code-vectors are selected in terms of minimum distortion. The search continues until the final stage, when only one best code-vector is determined.

[0008] In predictive quantization a target vector for quantization in the current frame is the mean-removed input vector minus a predictive value. The predicted value is the previous quantized vector multiplied by a known prediction matrix. In switched prediction, there is more than one possible prediction matrix and the best prediction matrix is selected for each frame. See S. Wang, et al., "Product Code Vector Quantization of LPC Parameters," in Speech and Audio Coding for Wireless and Network Applications," Ch. 31, pp. 251-258, Kluwer Academic Publishers, 1993.
[0009] It is highly desirable to provide an improved weighted distance measure that better correlates with subjective speech quality.

SUMMARY OF THE INVENTION

[0010] In accordance with a preferred embodiment the present invention provides an improved method of vector quantization of LSF transformation of LPC coefficients by a new weighted distance measure that better correlates with subjective speech quality. This weighting includes running samples from the LPC filter from an impulse and applying these samples to a perceptual weighting filter.

DESCRIPTION OF THE DRAWINGS

[0011] Embodiments of the present invention will now be further described, by way of example, with reference to the accompanying drawings in which:

Fig. 1 is a block diagram of Mixed Excitation Linear Prediction Coder;
Fig. 2 is a block diagram of switch-predictive vector quantization encoder according to the present invention;
Fig. 3 is a block diagram of a decoder according to the present invention;
Fig. 4 is a flow chart for determining a weighted distance measure in accordance with an embodiment of the present invention; and
Fig. 5 is a block diagram of an encoder according to an embodiment of the present invention.

DESCRIPTION OF PREFERRED EMBODIMENTS OF THE PRESENT INVENTION

[0012] The new quantization method, like the one used in the 2.4 kb/s Federal Standard MELP coder, uses multi-stage vector quantization (MSVQ) of the Line Spectral Frequency (LSF) transformation of the LPC coefficients (LeBlanc, et al., entitled "Efficient Search and Design Procedures for Robust Multi-Stage VQ or LPC Parameters for 4kb/s Speech Coding," IEEE Transactions on Speech and Audio Processing, Vol. 1, No. 4, October 1993, pp. 373-385.) An efficient codebook search for multi-stage VQ is disclosed in US Patent Application Serial No. 09/003,172 cited above. However, the method, described herein, improves on the previous one in two ways: the use of switched prediction to take advantage of time redundancy and the use of a new weighted distance measure that better correlates with subjective speech quality.
[0013] In the Federal Standard MELP coder, the input LSF vector is quantized directly using MSVQ. However, there is a significant redundancy between LSF vectors of neighboring frames, and quantization accuracy can be improved by exploiting this redundancy. As discussed previously in predictive quantization, the target vector for quantization in the current frame is the mean-removed input vector minus a predicted value, where the predicted value is the previous quantized vector multiplied by a known prediction matrix. In switched prediction, there is more than one possible prediction matrix, and the best predictor or prediction matrix is selected for each frame. In accordance with an embodiment of the present invention, both the predictor matrix and the MSVQ codebooks are switched. For each input frame, we search every possible predictor/codebooks set combination for the predictor/codebooks set which minimizes the squared error. An index corresponding to this pair and the MSVQ codebook indices are then encoded for transmission. This differs from previous techniques in that the codebooks are switched as well as the predictors. Traditional methods share a single codebook set in order to reduce codebook storage, but we have found that the MSVQ codebooks used in switched predictive quantization can be considerably smaller than non-predictive codebooks, and that multiple smaller codebooks do not require any more storage space than one larger codebook. From our experiments, the use of separate predictor/codebooks pairs results in a significant performance improvement over a single shared codebook, with no increase in bit rate.
[0014] Referring to the LSF encoder with switched predictive quantizer 20 of Fig. 2, the 10 LPC coefficients are transformed by transformer 23 to 10 LSF coefficients of the Line Spectral Frequency (LSF) vectors. The LSF has 10 dimensional elements or coefficients (for 10 order all-pole filter). The LSF input vector is subtracted in adder 22 by a selected mean vector and the mean-removed input vector is subtracted in adder 25 by a predicted value. The resulting target vector for quantization vector e in the current frame is applied to multi-stage vector quantizer (MSVQ) 27. The predicted value is the previous quantized vector multiplied by a known prediction matrix at multiplier 26. The predicted

value in switched prediction has more than one possible prediction matrix. The best predictor (prediction matrix and mean vector) is selected for each frame. In accordance with an embodiment of the present invention, both the predictor (the prediction matrix and mean vector) and the MSVQ codebook set are switched. A control 29 first switches in via switch 28 prediction matrix 1 and mean vector 1 and first set of codebooks 1 in quantizer 27. The index corresponding to this first prediction matrix and the MSVQ codebook indices for the first set of codebooks are then provided out of the quantizer to gate 37. The predicted value is added to the quantized output $\hat{e}$ for the target vector e at adder 31 to produce a quantized mean-removed vector. The mean-removed vector is added at Adder 70 to the selected mean vector to get quantized vector $\hat{X}$. The squared error for each dimension is determined at squarer 35. The weighted squared error between the input vector X_i and the delayed quantized vector $\hat{X}_i$ is stored at control 29. The control 29 applies control signals to switch in via switch 28 prediction matrix 2 and mean vector 2 and codebook 2 set to likewise measure the weighted squared error for this set at squarer 35. The measured error from the first pair of prediction matrix 1 (with mean vector 1) and codebooks set 1 is compared with prediction matrix 2 (with mean vector 2) and codebook set 2. The set of indices for the codebooks with the minimum error is gated at gate 37 out of the encoder as encoded transmission of indices and a bit is sent out at terminal 38 from control 29 indicating from which pair of prediction matrix and codebooks set the indices was sent (codebook set 1 with mean vector 1 and predictor matrix 1 or codebook set 2 and prediction matrix 2 with mean vector 2). The mean-removed quantized vector from adder 31 associated with the minimum error is gated at gate 33a to frame delay 33 so as to provide the previous mean-removed quantized vector to multiplier 26.

[0015] Fig. 3 illustrates a decoder 40 for use with LSF encoder 20. At the decoder 40, the indices for the codebooks from the encoding are received at the quantizer 44 with two sets of codebooks corresponding to codebook set 1 and 2 in the encoder. The bit from terminal 38 selects the appropriate codebook set used in the encoder. The LSF quantized input is added to the predicted value at adder 41 where the predicted value is the previous mean-removed quantized value (from delay 43) multiplied at multiplier 45 by the prediction matrix at 42 that matches the best one selected at the encoder to get mean-removed quantized vector. Both prediction matrix 1 and mean value 1 and prediction matrix 2 and mean value 2 are stored at storage 42 of the decoder. The 1 bit from terminal 38 of the encoder selects the prediction matrix and the mean value at storage 42 that matches the encoder prediction matrix and mean value. The quantized mean-removed vector is added to the selected mean value at adder 48 to get the quantized LSF vector. The quantized LSF vector is transformed to LPC coefficients by transformer 46.

[0016] As discussed previously, LSF vector coefficients correspond to the LPC coefficients. The LSF vector coefficients have better quantization properties than LPC coefficients. There is a 1 to 1 transformation between these two vector coefficients. A weighting function is applied for a particular set of LSFs for a particular set of LPC coefficients that correspond.

[0017] The Federal Standard MELP coder uses a weighted Euclidean distance for LSF quantization due to its computational simplicity. However, this distance in the LSF domain does not necessarily correspond well with the ideal measure of quantization accuracy: perceived quality of the processed speech signal. The applicant has previously shown in the paper on the new 2.4 kb/s Federal Standard that a perceptually-weighted form of log spectral distortion has close correlation with subjective speech quality. The applicant teaches herein in accordance with an embodiment a weighted LSF distance which corresponds closely to this spectral distortion. This weighting function requires looking into the details of this transformation for a particular set of LSFs for a particular input vector x which is a set of LSFs for a particular set of LPC coefficients that correspond to that set. The coder computes the LPC coefficients and as discussed above, for purposes of quantization, this is converted to LSF vectors which are better behaved. As shown in Fig. 1, the actual synthesizer will take the quantized vector $\hat{X}$ and perform an inverse transformation to get an LPC filter for use in the actual speech synthesis. The optimal LSF weights for un-weighted spectral distortion are computed using the formula presented in paper of Gardner, et al., entitled, "Theoretical Analysis of the High-Rate Vector Quantization of the LPC Parameters," IEEE Transactions on Speech and Audio Processing, Vol. 3, No. 5, September 1995, pp. 367-381.

$$W_i = R_A(0)R_i(0) + 2\sum_{m=1}^{p-1} R_A(m)R_i(m)$$

where $R_A(m)$ is the autocorrelation of the impulse response of the LPC synthesis filter at lag m, and $R_i(m)$ is the correlation of the elements in the ith column of the Jacobian matrix of the transformation from LSF's to LPC coefficients. Therefore for a particular input vector x we compute the weight W_i.

[0018] The difference in the present solution is that perceptual weighting is applied to the synthesis filter impulse response prior to computation of the autocorrelation function $R_A(m)$, so as to reflect a perceptually-weighted form of spectral distortion.

[0019] In accordance with the weighting function as applies to the embodiment of Fig. 2, the weighting W_i is applied to the squared error at 35. The weighted output from error detector 35 is:

$$\sum W_i(X_i - \hat{X}_i)^2 .$$

Each entry in a 10 dimensional vector has a weight value. The error sums the weight value for each element. In applying the weight, for example, one of the elements has a weight value of three and the others are one then the element with three is given an emphasis by a factor of three times that of the other elements in determining error.

[0020] As stated previously, the weighting function requires looking into the details of the LPC to LSF conversion. The weight values are determined by applying an impulse to the LPC synthesis filter 21 and providing the resultant sampled output of the LPC synthesis filter 21 to a perceptual weighting filter 47. A computer 39 is programmed with a code based on a pseudo code that follows and is illustrated in the flow chart of Fig. 4. An impulse is gated to the LPC filter 21 and N samples of LPC synthesis filter response (step 51) are taken and applied to a perceptual weighting filter 37 (step 52). In accordance with one embodiment of the invention low frequencies are weighted more than high frequencies and use the well known Bark scale which matches how the human ear responds to sounds. The equation for Bark weighting $W_B(f)$ is

$$W_B(f) = \frac{1}{25 + 75(1 + 1.4(\frac{f}{1000})^2)^{0.69}}$$

The coefficients of a filter with this response are determined in advance and stored and time domain coefficients are stored. An 8 order all-pole fit to this spectrum is determined and these 8 coefficients are used as the perceptual weighting filter. The following steps follow the equation for un-weighted spectral distortion from Gardner, et al. paper found on page 375 are expressed as

$$W_i = R_A(0)R_i(0) + 2\sum_{m=1}^{p-1} R_A(m)R_i(m)$$

where $R_A(m)$ is the autocorrelation of the impulse response of the LPC synthesis filter at lag m, where

$$R_A(k) = \sum_{n=0}^{a} h(n)h(n+k)$$

h(n) is an impulse response, $R_i(m)$ is

$$R_{i}(m) = \sum_{n=1}^{v-m} (J_\omega(\omega))_{n,i}\,(J_\omega(\omega))_{m+n,i}$$

$$= \sum_{n=1}^{v-m} j_i(n)\,j_i(m+n)$$

is the correlation function of the elements in the ith column of the Jacobian matrix $J_\omega(\omega)$ of the transformation from LSFs to LPC coefficients. Each column of $J_\omega(\omega)$ can be found by

$$= \begin{cases} \sin(\omega_i)e^{-j\omega} \prod\limits_{j=0;\, j\neq(i+1)/2}^{v/2} \tilde{p}_j(\omega); i \text{ odd} \\ \sin(\omega_i)e^{-j\omega} \prod\limits_{j=0;\, j\neq i/2}^{v/2} \tilde{q}_j(\omega); i \text{ even} \end{cases}$$

since

$$\prod_{j=0;j\neq i}^{v/2} \tilde{p}_j(\omega) = P(\omega) / \tilde{p}_i(\omega)$$

The values of $j_i(n)$ can be found by simple polynomial division of the coefficients of $P(\omega)$ by the coefficients of $\tilde{p}_i(\omega)$. Since the first coefficient of $\tilde{p}_i(\omega) = 1$, no actual divisions are necessary in this procedure. Also, $j_i(n) = j_i(v + 1 - n)$; i odd; $0 < n < v$, so only half the values must be computed. Similar conditions with an anti-symmetry property exist for the even columns.

[0021] The autocorrelation function of the weighted impulse response is calculated (step 53 in Fig. 4). From that the Jacobian matrix for LSFs is computed (step 54). The correlation of rows of Jacobian matrix is then computed (step 55). The LSF weights are then calculated by multiplying correlation matrices (step 56). The computed weight value from computer 39, in Fig. 2, is applied to the error detector 35. The indices from the prediction matrix/codebook set with the least error is then gated from the quantizer 27. The system may be implemented using a microprocessor encapsulating computer 39 and control 29 utilizing the following pseudo code. The pseudo code for computing the weighting vector from the current LPC and LSF follows:

```
/* Compute weighting vector from current LPC and LSF's */
Compute N samples of LPC synthesis filter impulse response
Filter impulse response with perceptual weighting filter
Calculate the autocorrelation function of the weighted impulse response
Compute Jacobian matrix for LSF's
Compute correlation of rows of Jacobian matrix
Calculate LSF weights by multiplying correlation matrices
```

[0022] The code for the above is provided in Appendix A.

The pseudo code for the encode input vector follows:

```
/* Encode input vector */
For all predictor, codebook pairs
    Remove mean from input LSF vector
    Subtract predicted value to get target vector
    Search MSVQ codebooks for best match to target vector using weighted distance
    If Error <Emin
        Emin = Error
        best predictor index = current predictor
    Endif
End
Endcode best predictor index and codebook indices for transmission
```

[0023] The pseudo code for regenerate quantized vector follows:

```
/* Regenerate quantized vector */
Sum MSVQ codevectors to produce quantized target
Add predicted value
Update memory of past quantized values (mean-removed)
Add mean to produce quantized LSF vector
```

[0024] We have implemented a 20-bit LSF quantizer based on this new approach which produces equivalent performance to the 25-bit quantizer used in the Federal Standard MELP coder, at a lower bit rate. There are two predictor/codebook pairs, with each consisting of a diagonal first-order prediction matrix and a four stage MSVQ with codebook of size 64, 32, 16, and 16 vectors each. Both the codebook storage and computational complexity of this new quantizer are less than in the previous version.

[0025] Although the present invention and its advantages have been described in detail, it should be understood

that various changes, substitutions and alterations can be made herein without departing from the spirit and scope of the invention

[0026] For example it is anticipated that the system and method be used without switched prediction for each frame as illustrated in Fig. 5 wherein the weighted error for each frame would be determined at error detector and codebook indices with the least error would be gated out by control 29 and gate 37. For each frame, the LPC filtered samples of the impulse at filter 21 should be filtered by perception weighting filter 47 and processed by computer 39 using code such as described in the pseudo code to provide the weight vales. Also the perception weighting filter may use other perceptual weighting besides the bark scale that is perceptually motivated such as weighting low frequencies more than high frequencies, or the perceptual weighting filter as is presently used in CELP coders.

[0027] The scope of the present disclosure includes any novel feature or combination of features disclosed therein either explicitly or implicitly or any generalisation thereof irrespective of whether or not it relates to the claimed invention or mitigates any or all of the problems addressed by the present invention. The applicant hereby gives notice that new claims may be formulated to such features during the prosecution of this application or of any such further application derived therefrom. In particular, with reference to the appended claims, features from dependent claims may be combined with those of the independent claims and features from respective independent claims may be combined in any appropriate manner and not merely in the specific combinations enumerated in the claims.

APPENDIX A

```
/*  Function vq_lspw: compute LSP weights

   Inputs:
     *p_lsp - LSP array
     *pc - LPC coefficients
     p - LPC model order

   Output:
     *w - array of weights

   Copyright 1997, Texas Instruments

*/

Float *vq_lspw(Float *w,Float *p_lsp,Float *pc,Int p)
{
     Int i,j,k,m;
     Float d, tmp, *tp, *ir, *R, *pz, *qz, *rem, *t, **J, **RJ;
     static Float bark_wt[8] = {
          -0.84602182,
          0.27673657,
          -0.10480262,
          0.05609138,
          -0.03315923,
          0.02132074,
          -0.01359822,
          0.00398910,
     };

     /* Allocate local array memory */
     MEM_ALLOC(MALLOC,ir,IRLENGTH+p,Float);
     ir=&ir[p];
     MEM_ALLOC(MALLOC,R,p,Float);
     MEM_ALLOC(MALLOC,pz,p+2,Float);
     MEM_ALLOC(MALLOC,qz,p+2,Float);
     MEM_ALLOC(MALLOC,rem,p+2,Float);
     MEM_ALLOC(MALLOC,t,3,Float);
     MEM_2ALLOC(MALLOC,J,p+1,p+1,Float);
     MEM_2ALLOC(MALLOC,RJ,p+1,p,Float);

     /* calculate IRLENGTH samples of the synthesis filter impulse response*/
     for (i=-p; i<IRLENGTH; i++)
          ir[i] = 0.0;
     ir[0] = 1.0;
     for (i=0; i<IRLENGTH; i++)
     {
          for (j=1; j<=p; j++)
               ir[i] -= pc[j] * ir[i-j];
     }

     /* use all-pole model for frequency weighting */
     for (i=0; i<IRLENGTH; i++)
     {
          for (j=1; j<=8; j++)
               ir[i] -= bark_wt[j-1] * ir[i-j];
     }

     /* calculate the autocorrelation function of the impulse response */
     for (m=0; m<p; m++)                              /* for lags of 0 to p-1 */
     {
          R[m] = 0.0f;
          for (i=0; i<IRLENGTH-m; i++)
               R[m] += ir[i] * ir[i+m];
     }

     /* calculate P(z) and Q(z) */
     for (i=1; i<=p; i++)
     {
          pz[i] = pc[i] + pc[p+1-i];
          qz[i] = pc[i] - pc[p+1-i];
     }
```

APPENDIX A

```
pz[0] = qz[0] = pz[p+1] = 1.0f;
qz[p+1] = -1.0f;

/* calculate the J matrix */
/* use the rows of J to store the polynomials */
/* (rather than the columns, as in Gardner) */
t[0] = t[2] = 1.0f;
for (i=1; i<=p; i++)                                    /* for all the rows of J */
{
    t[1] = -2.0f * cos(PI * p_lsp[i]);
    tmp = sin(PI * p_lsp[i]);
    if (i != 2 * (i/2)) tp = pz;                    /* i is odd;  use p(z) */
    else               tp = qz;                     /* i is even; use q(z) */
    /* divide polynomial tp by polynomial t and put the result into */
    /* row J[i] */
    for (j=0; j<=p+1; j++)
        rem[j] = tp[j];
    for (k=p; k>=1; k--)
    {
        J[i][k] = rem[k+1];
        for (j=k; j>=k-1; j--)
            rem[j] -= J[i][k] * t[j-k+1];
    }
    /* multiply the ith row by the sin() term */
    for (j=1; j<=p; j++)
        J[i][j] *= tmp;
}

/* determine the 'correlation' function of the rows of J */
for (i=1; i<=p; i++)                                    /* for each row */
{
    for (m=0; m<p; m++)                             /* for each lag */
    {
        RJ[i][m] = 0.0f;
        /* for each element in the row */
        for (j=1; j<=p-m; j++)
            RJ[i][m] += J[i][j] * J[i][j+m];
    }
}
/* finish the weight calculation */
for (i=1; i<=p; i++)
{
    tmp = 0.0f;
    for (m=1; m<p; m++)
        tmp += R[m] * RJ[i][m];
    w[i-1] = R[0] * RJ[i][0] + 2.0f * tmp;

}

/* Free local memory */
ir=&ir[-p];
MEM_FREE(FREE,ir);
MEM_FREE(FREE,R);
MEM_FREE(FREE,pz);
MEM_FREE(FREE,qz);
MEM_FREE(FREE,rem);
MEM_FREE(FREE,t);
MEM_2FREE(FREE,J);
MEM_2FREE(FREE,RJ);

return(w);
```

Claims

1. A method of vector quantization of LPC coefficients comprising the steps of:

   translating LPC coefficients to LSF coefficients;
   providing a quantizer with a codebook for quantizing LSF target vectors;
   searching within said codebook for determining LSF target vectors that result in quantized output that best match LPC coefficients;
   applying said target vectors to said codebook to get quantized vectors;
   said searching step comprising the step of determining the squared error multiplied by a weighting value for each dimension between the LSF coefficients and the quantized output wherein said weighting value is a function of perceptual weighting;
   and said determining step including the steps of:
   calculating an autocorrelation function of a weighted impulse response;
   computing a Jacobian matrix for said LSF vectors;
   computing the correlation of rows of the Jacobian matrix; and
   calculating LSF weights by multiplying correlation matrices.

2. The method of Claim 1 wherein said determining step comprises the further steps for finding said weighting value of:

   applying an impulse to said LPC filter and running N samples of the LPC synthesis response; and
   filtering the samples with a perceptual filter;
   calculating the autocorrelation function of the weighted impulse response;
   computing the Jacobian matrix for said LSF vectors;
   computing the correlation of rows of Jacobian matrix; and
   calculating LSF weights by multiplying correlation matrices.

3. The method of Claim 2 wherein the step of filtering the samples with said perceptual filter comprises weighting low frequencies more than high frequencies.

4. The method of Claim 3 wherein the step of filtering the samples with said perceptual filter comprises following the bark scale.

5. The method of any preceding Claim wherein said step of providing said quantizer comprises providing a multistage vector quantizer.

6. The method of any preceding Claim wherein said step of providing said quantizer comprises providing a quantizer having one or more sets of codebooks.

7. A quantizer for a coder including an LPC filter and a translator for translating LPC coefficients to LSF coefficients comprising:

   a codebook responsive to said LSF target vector for quantizing LSF target vectors;
   means for searching within said codebooks for determining LSF target vectors that result in quantized output that best match LPC coefficients;
   means for applying said LSF target vectors to said codebook to provide a quantized output;
   said searching means including means for applying an impulse to said LPC filter;
   means for running samples of said LPC response;
   a perceptual filter for filtering said samples; and
   means for calculating an autocorrelation function by weighted response, a Jacobian matrix for said LSF vectors, a correlation of rows of Jacobian matrix, and LSF weights by multiplying correlation matrices.

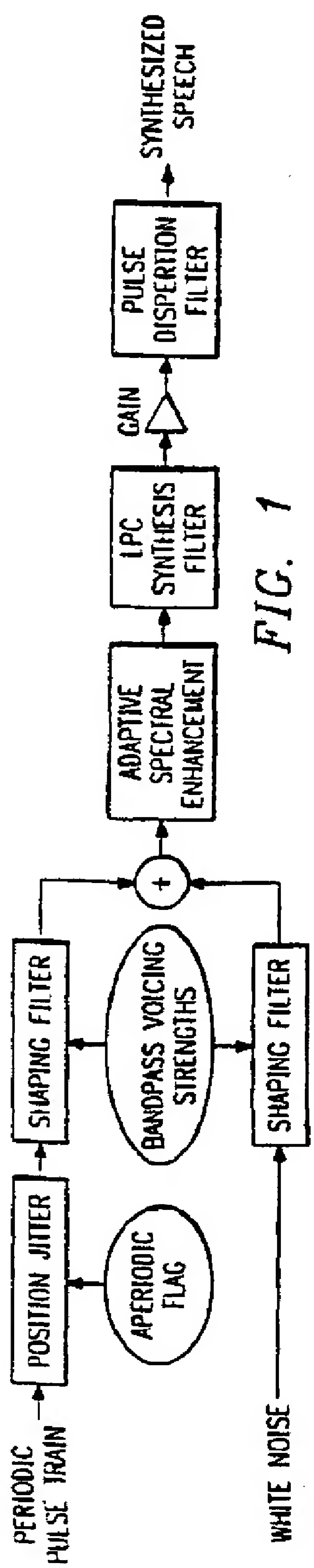

PERIODIC PULSE TRAIN
POSITION JITTER
APERIODIC FLAG
SHAPING FILTER
BANDPASS VOICING STRENGTHS
SHAPING FILTER
WHITE NOISE
+
ADAPTIVE SPECTRAL ENHANCEMENT
LPC SYNTHESIS FILTER
GAIN
PULSE DISPERTION FILTER
SYNTHESIZED SPEECH

FIG. 1

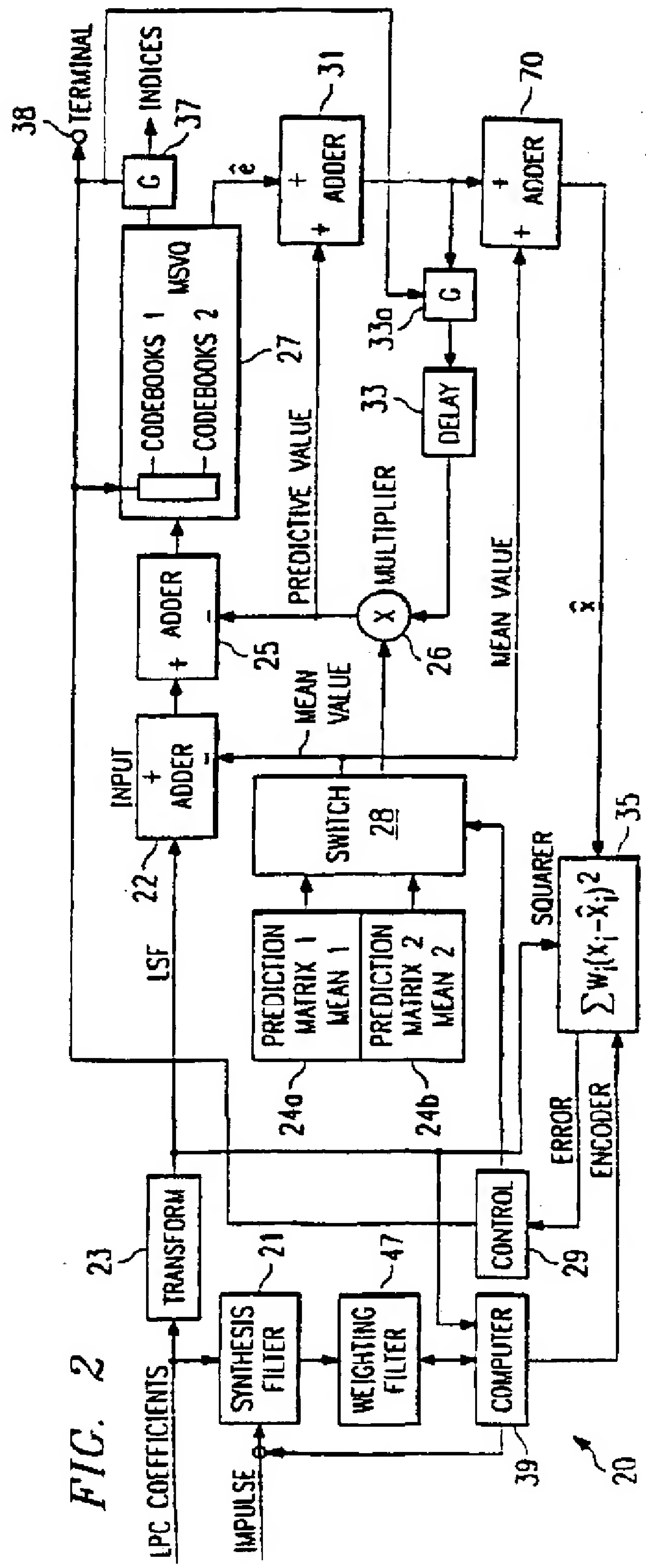

LPC COEFFICIENTS
IMPULSE
SYNTHESIS FILTER
21
WEIGHTING FILTER
47
COMPUTER
39
20
23
TRANSFORM
CONTROL
29
LSF
24a
24b
PREDICTION MATRIX 1 MEAN 1
PREDICTION MATRIX 2 MEAN 2
SWITCH
28
22
INPUT
+
ADDER
−
MEAN VALUE
+
ADDER
−
25
X
MULTIPLIER
26
PREDICTIVE VALUE
MEAN VALUE
CODEBOOKS 1
CODEBOOKS 2
MSVQ
27
G
37
INDICES
38
TERMINAL
ê
+
+
ADDER
31
33
DELAY
33a
G
+
+
ADDER
70
x̂
SQUARER
$\sum w_i(x_i-\hat{x}_i)^2$
35
ERROR
ENCODER

FIG. 2

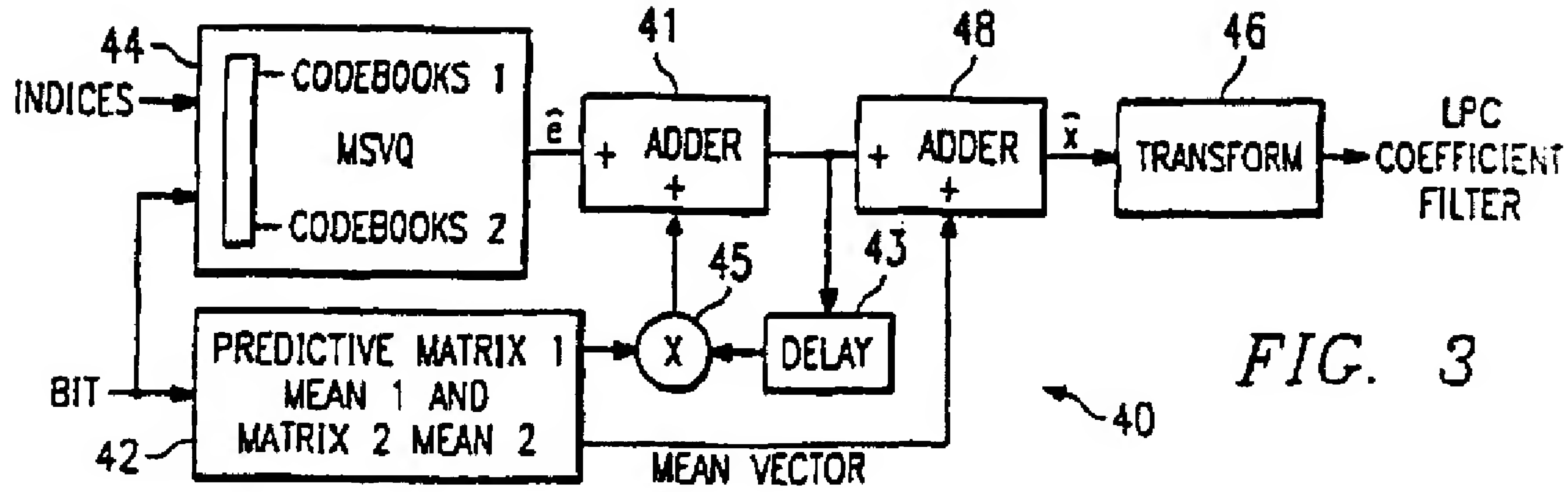


FIG. 3

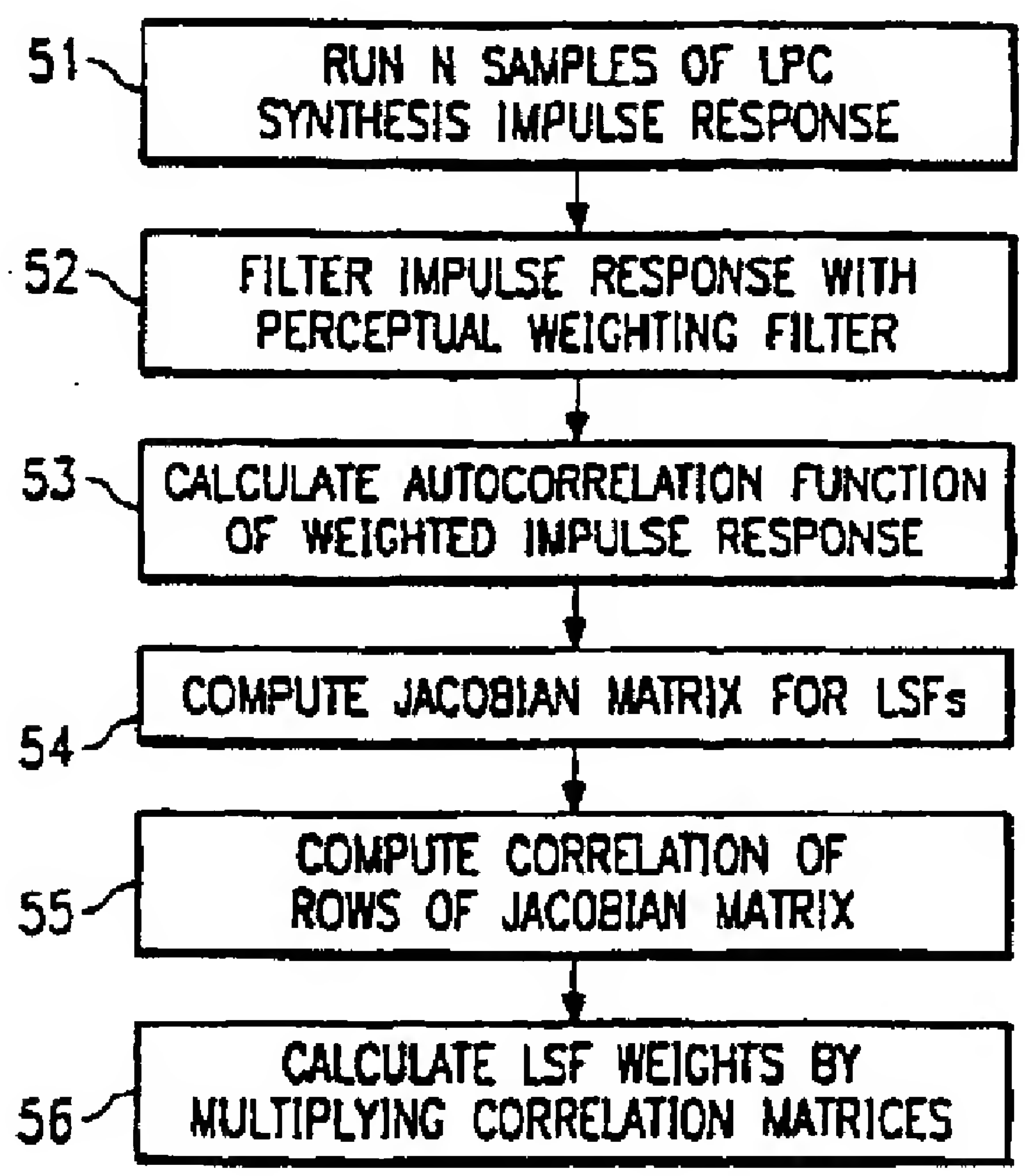


FIG. 4

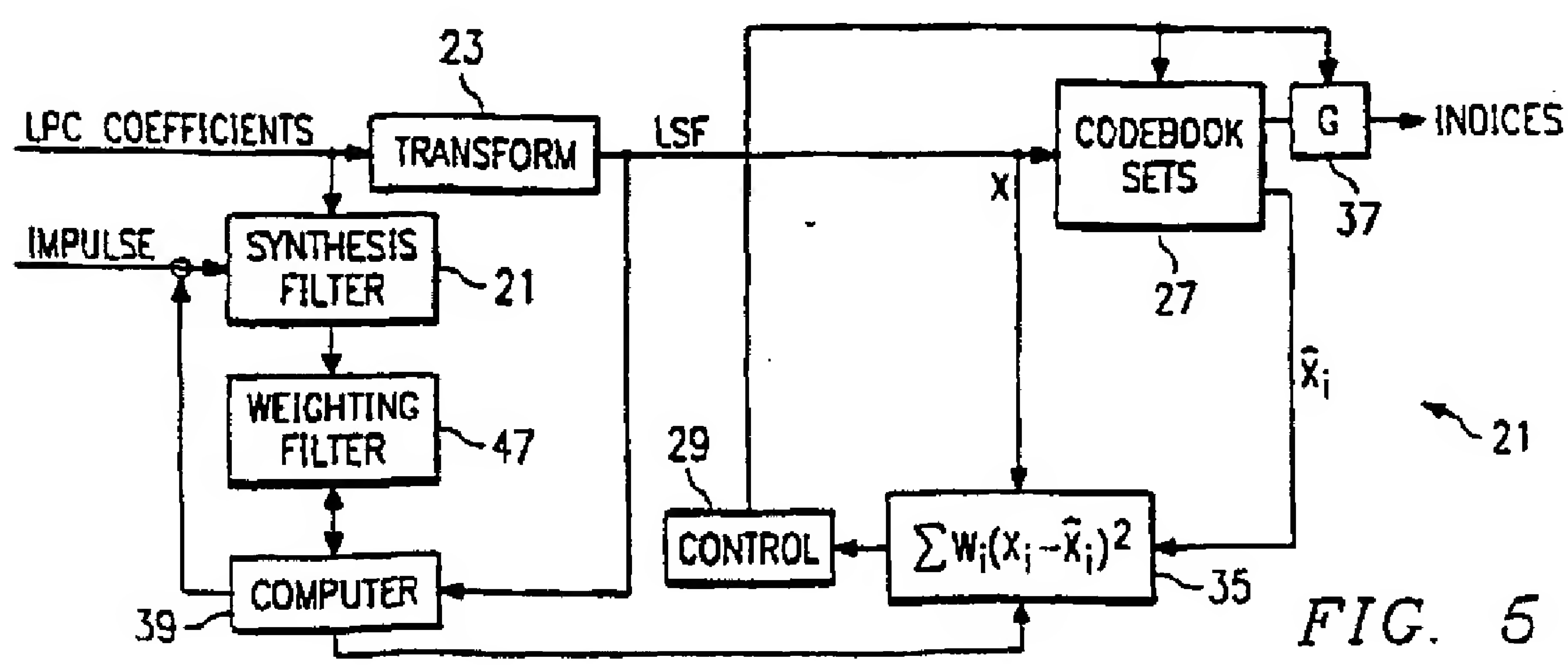


FIG. 5

(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) EP 0 899 720 A3

(12) **EUROPEAN PATENT APPLICATION**

(88) Date of publication A3:
15.09.1999 Bulletin 1999/37

(51) Int Cl.6: **G10L 9/14**, G10L 3/02

(43) Date of publication A2:
03.03.1999 Bulletin 1999/09

(21) Application number: **98306906.3**

(22) Date of filing: **27.08.1998**

(84) Designated Contracting States:
AT BE CH CY DE DK ES FI FR GB GR IE IT LI LU MC NL PT SE
Designated Extension States:
AL LT LV MK RO SI

(30) Priority: **28.08.1997 US 57114 P**

(71) Applicant: **TEXAS INSTRUMENTS INC.**
Dallas, Texas 75243 (US)

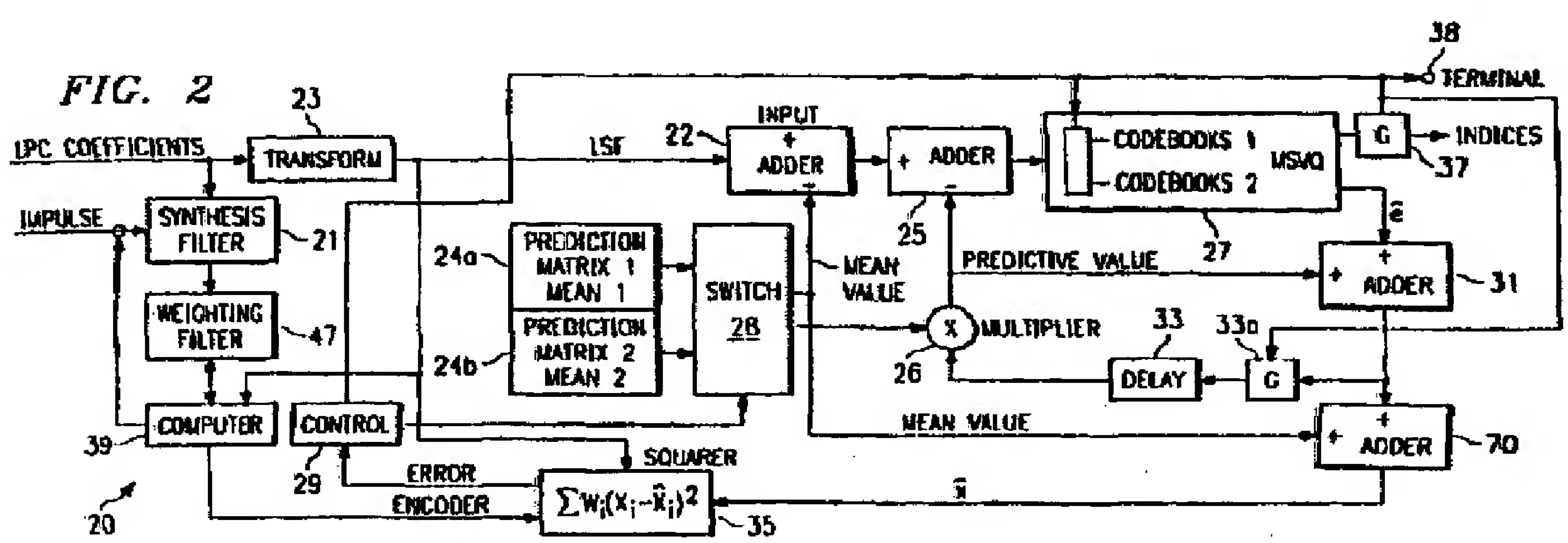
(72) Inventor: **McCree, Alan V.**
Dallas Texas 75248 (US)

(74) Representative: **Potter, Julian Mark et al**
D. Young & Co.,
21 New Fetter Lane
London EC4A 1DA (GB)

(54) **Quantization of linear prediction coefficients**

(57) A new method for quantization of the LPC coefficients in a speech coder includes a new weighted error measure including every frame sampling an impulse response from LPC filter (21) of said coder, filtering the samples using a perceptual weighting filter (47) and processing in a computer (39) to calculate autocorrelation function of the weighted impulse response, computing Jacobian matrix for LSF (Line Spectral Frequency), computing correlation of rows of Jacobian matrix and calculating LSF weights by multiplying correlation matrices.

FIG. 2



EP 0 899 720 A3

European Patent Office

EUROPEAN SEARCH REPORT

Application Number
EP 98 30 6906

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int.Cl.6)
P,X	MCCREE A ET AL: "A 1.7 kb/s MELP coder with improved analysis and quantization" PROCEEDINGS OF THE 1998 IEEE INTERNATIONAL CONFERENCE ON ACOUSTICS, SPEECH AND SIGNAL PROCESSING, ICASSP '98 (CAT. NO.98CH36181), PROCEEDINGS OF THE 1998 IEEE INTERNATIONAL CONFERENCE ON ACOUSTICS, SPEECH AND SIGNAL PROCESSING, SEATTLE, WA, USA, 12-1, pages 593-596 vol.2, XP002109401 1998, New York, NY, USA, IEEE, USAISBN: 0-7803-4428-6 * paragraph [2.2.1] * ---	1-7	G10L9/14 G10L3/02
E	EP 0 905 680 A (TEXAS INSTRUMENTS INC) 31 March 1999 (1999-03-31) * page 5 * ---	1-7	
D,A	GARDNER W R ET AL: "THEORETICAL ANALYSIS OF THE HIGH-RATE VECTOR QUANTIZATION OF PLC PARAMETERS" IEEE TRANSACTIONS ON SPEECH AND AUDIO PROCESSING, vol. 3, no. 5, 1 September 1995 (1995-09-01), pages 367-381, XP000641961 ISSN: 1063-6676 * paragraph [III.D] * * abstract * --- -/--	1-7	TECHNICAL FIELDS SEARCHED (Int.Cl.6) G10L

The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
THE HAGUE	16 July 1999	Krembel, L

CATEGORY OF CITED DOCUMENTS

X : particularly relevant if taken alone
Y : particularly relevant if combined with another document of the same category
A : technological background
O : non-written disclosure
P : intermediate document

T : theory or principle underlying the invention
E : earlier patent document, but published on, or after the filing date
D : document cited in the application
L : document cited for other reasons

& : member of the same patent family, corresponding document

European Patent Office

EUROPEAN SEARCH REPORT

Application Number
EP 98 30 6906

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int.Cl.6)
A	MCCREE A ET AL: "A 2.4 kbit/s MELP coder candidate for the new U.S. Federal Standard" 1996 IEEE INTERNATIONAL CONFERENCE ON ACOUSTICS, SPEECH, AND SIGNAL PROCESSING CONFERENCE PROCEEDINGS (CAT. NO.96CH35903), 1996 IEEE INTERNATIONAL CONFERENCE ON ACOUSTICS, SPEECH, AND SIGNAL PROCESSING CONFERENCE PROCEEDINGS, ATLANTA, GA, USA, 7-10 M, pages 200-203 vol. 1, XP002109402 1996, New York, NY, USA, IEEE, USAISBN: 0-7803-3192-3 * paragraph [3.1.2] * ---	1-7	
A	COHN R P ET AL: "Incorporating perception into LSF quantization some experiments" 1997 IEEE INTERNATIONAL CONFERENCE ON ACOUSTICS, SPEECH, AND SIGNAL PROCESSING (CAT. NO.97CB36052), 1997 IEEE INTERNATIONAL CONFERENCE ON ACOUSTICS, SPEECH, AND SIGNAL PROCESSING, MUNICH, GERMANY, 21-24 APRIL 1997, pages 1347-1350 vol.2, XP002109403 1997, Los Alamitos, CA, USA, IEEE Comput. Soc. Press, USAISBN: 0-8186-7919-0 * paragraph [01.1] * * paragraph [0003] * ---	1-7	
	-/--		TECHNICAL FIELDS SEARCHED (Int.Cl.6)

The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
THE HAGUE	16 July 1999	Krembel, L

CATEGORY OF CITED DOCUMENTS

X : particularly relevant if taken alone
Y : particularly relevant if combined with another document of the same category
A : technological background
O : non-written disclosure
P : intermediate document

T : theory or principle underlying the invention
E : earlier patent document, but published on, or after the filing date
D : document cited in the application
L : document cited for other reasons

& : member of the same patent family, corresponding document

EPO FORM 1503 03.82 (P04C01)

European Patent Office

EUROPEAN SEARCH REPORT

Application Number
EP 98 30 6906

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int.Cl.6)
A	PALIWAL K K ET AL: "EFFICIENT VECTOR QUANTIZATION OF LPC PARAMETERS AT 24 BITS/FRAME" SPEECH PROCESSING 1, TORONTO, MAY 14 - 17, 1991, vol. 1, no. CONF. 16, 14 May 1991 (1991-05-14), pages 661-664, XP000245315 INSTITUTE OF ELECTRICAL AND ELECTRONICS ENGINEERSISBN: 0-7803-0003-3 * paragraph [0003] * -----	1,7	
			TECHNICAL FIELDS SEARCHED (Int.Cl.6)

The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
THE HAGUE	16 July 1999	Krembel, L

CATEGORY OF CITED DOCUMENTS

X : particularly relevant if taken alone
Y : particularly relevant if combined with another document of the same category
A : technological background
O : non-written disclosure
P : intermediate document

T : theory or principle underlying the invention
E : earlier patent document, but published on, or after the filing date
D : document cited in the application
L : document cited for other reasons

& : member of the same patent family, corresponding document

EPO FORM 1503 03.82 (P04C01)

ANNEX TO THE EUROPEAN SEARCH REPORT ON EUROPEAN PATENT APPLICATION NO. EP 98 30 6906

This annex lists the patent family members relating to the patent documents cited in the above-mentioned European search report.
The members are as contained in the European Patent Office EDP file on
The European Patent Office is in no way liable for these particulars which are merely given for the purpose of information.

16-07-1999

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
EP 0905680 A	31-03-1999	NONE	

EPO FORM P0459

For more details about this annex : see Official Journal of the European Patent Office, No. 12/82